Claims:

1. A hearing-aid system for processing an acoustic input signal and providing at least one output acoustic signal to a user of the hearing-aid system, the hearing-aid system comprising a first channel and a second channel, wherein one of the channels includes an adaptive delay and the first channel includes:

a) a first directional unit for receiving the acoustic input signal and providing a first directional signal;

b) a first correlative unit coupled to the first directional unit for receiving the first directional signal and providing a first noise reduced signal by utilizing correlative measures for identifying a speech signal of interest in the first directional signal; and,

c) a first compensator coupled to the first correlative unit for receiving the first noise reduced signal and providing a first compensated signal for compensating for a hearing loss of the user.

2. The hearing-aid system of claim 1, wherein the second channel includes:

d) a second directional unit for receiving the acoustic input signal and providing a second directional signal;

e) a second correlative unit coupled to the second directional unit for receiving the second directional signal and providing a second noise reduced signal by utilizing correlative measures for identifying a speech signal of interest in the second directional signal; and,

f) a second compensator coupled to the second correlative unit for receiving the second noise reduced signal and providing a second compensated signal for compensating for a hearing loss of the user.

3. The hearing-aid system of claim 2, wherein the adaptive delay provides an appropriate delay to one of the first compensated signal and the second compensated signal for matching processing delay in the first and second channels.

4. The hearing-aid system of claim 1, wherein the correlative measures are provided by atomic decomposition phonemic processing.

5. The hearing-aid system of claim 4, wherein the atomic decomposition phonemic processing comprises mapping a portion of the first directional signal into a five-dimensional space which comprises dimensions of: duration in time, duration in frequency, temporal centers of gravity, spectral centers of gravity, and change of spectral centers of gravity.

6. The hearing-aid system of claim 5, wherein the mapping is performed according to:

$$h_{T_c,F_c,\sigma_T,\sigma_F,\beta}(t,f)=\frac{1}{2\pi\sigma_T^2\sigma_F^2}\,e^{-\left[\frac{1}{2(1-\beta^2)}\left(\frac{(t-T_c)^2}{\sigma_T^2}-\frac{2\beta(t-T_c)(f-F_c)}{\sigma_T\sigma_F}+\frac{(f-F_c)^2}{\sigma_F^2}\right)\right]}.$$

7. The hearing-aid system of claim 4, wherein the atomic decomposition phonemic processing comprises correlating an atom with a portion of the first directional signal according to: $\gamma_p = \arg\max_{\gamma}\left|\left\langle s_{p-1}(t), f\left(\sigma_T,\sigma_F\right)h_\gamma(t)\right\rangle\right|^2$.

8. The hearing-aid system of claim 1, wherein the correlative measures are provided by acoustic correlative tracking and the first correlative unit comprises:

d) a correlator generator for receiving an input signal and generating a plurality of speech and environment correlates;

e) a control unit coupled to the correlator generator for receiving the speech correlates and the environment correlates and generating a control signal; and,

f) a processing unit coupled to the correlator generator and the control unit, the processing unit receiving the input signal, the speech correlates and the control signal and processing the speech correlates according to the control signal for extracting speech from the input signal.

9. The hearing-aid system of claim 8, wherein the processing unit processes the input signal by selecting appropriate speech correlates based on the environmental correlates and tracking the appropriate speech correlates.

10. The hearing-aid system of claim 9, wherein the processing unit employs one of a Kalman filter and a particle filter for tracking the appropriate speech correlates.

11. The hearing-aid system of claim 1, wherein the first compensator comprises:

d) a normal hearing model unit for receiving an input signal and generating a normal hearing signal;

e) a neuro-compensator unit for receiving the input signal and providing a pre-processed signal by applying a set of weights to the input signal;

f) a damaged hearing model unit connected to the neuro-compensator unit for receiving the pre-processed signal and providing an impaired hearing signal; and,

g) a comparison unit connected to the normal hearing model unit and the damaged hearing model unit for generating an error signal based on a comparison of the normal hearing signal and the impaired hearing signal; wherein, the error signal is provided to the neuro-compensator unit for adjusting the set of weights such that the normal hearing signal and the impaired hearing signal are substantially similar.

12. The hearing-aid system of claim 11, wherein the neuro-compensator is a neural network.

13. The hearing-aid system of claim 12, wherein the neuro-compensator applies a set of gain coefficients to the input signal, each gain coefficient being defined for a particular frequency band i according to $G_i = \dfrac{v_i f_i^2}{\sum_j w_j f_j^2 + \sigma}$

where f_i^2 is energy at frequency band i, w_j is a weight at frequency band j and σ is a constant related to the energy f_i^2.

14. The hearing-aid system of claim 12, wherein a weight W_j from the set of weights is defined for a particular time-slice at the i^{th} frequency band according to $W_i = \frac{v_i}{\left(\sum_{j=1}^{20} w_{ij} f_j\right)^{1/4} + \left[\sum_{k=0}^{4}\left(z_{ik} \sum_{j=1}^{20} f_j^{n-k}\right)^{1/4}\right] + \sigma}$ where f_j is the magnitude of the input signal in the j^{th} frequency band, v_i is optimized average gain, w_{ii} is optimized band to band inhibition, z_{ik} is optimized total power inhibition for past times and σ is a constant.

15. The hearing-aid system of claim 11, wherein the error signal is defined according to a Neural Articulation Index (NAI) of the form $NAI = \sum_{i=1}^{N} \alpha_i \cdot ND_i$ where N is a number of frequency bands, α_i is a weight for frequency band i, and ND (Neural Distortion) is defined by $ND = 1 - \frac{Test \cdot Control'}{Control \cdot Control'}$ where Test is a vector of instantaneous spiking rates provided by the damaged hearing model unit and Control is a vector of instantaneous spiking rates provided by the normal hearing model unit.

16. A noise reduction unit for use in a hearing aid, the noise reduction unit receiving an input signal and providing a noise reduced signal, wherein the noise reduction unit includes a correlative portion for providing correlative measures for identifying a speech signal of interest in the input signal and a tracking portion for tracking the speech signal of interest to produce the noise reduced signal.

17. The noise reduction unit of claim 16, wherein the correlative unit employs atomic decomposition phonemic processing to produce the correlative measures.

18. The noise reduction unit of claim 17, wherein the atomic decomposition phonemic processing comprises mapping a portion of the first directional signal into a five-dimensional space which comprises dimensions of: duration in time, duration in frequency, temporal centers of gravity, spectral centers of gravity, and change of spectral centers of gravity.

19. The noise reduction unit of claim 18, wherein the mapping is performed according to:

$$h_{T_c,F_c,\sigma_T,\sigma_F,\beta}(t,f)=\frac{1}{2\pi\sigma_T^2\sigma_F^2}e^{-\left[\frac{1}{2(1-\beta^2)}\left(\frac{(t-T_c)^2}{\sigma_T^2}-\frac{2\beta(t-T_c)(f-F_c)}{\sigma_T\sigma_F}+\frac{(f-F_c)^2}{\sigma_F^2}\right)\right]}.$$

20. The noise reduction unit of claim 17, wherein the atomic decomposition phonemic processing comprises correlating an atom with a portion of the first directional signal according to: $\gamma_p=\arg\max_{\gamma}\left|\left\langle s_{p-1}(t), f\left(\sigma_T,\sigma_F\right)h_{\gamma}(t)\right\rangle\right|^2$.

21. The noise reduction unit of claim 16, wherein the correlative measures are provided by acoustic correlative tracking and the noise reduction unit comprises:

a) a correlator generator for receiving the input signal and generating a plurality of speech and environment correlates;

b) a control unit coupled to the correlator generator for receiving the speech correlates and the environment correlates and generating a control signal; and,

c) a processing unit coupled to the correlator generator and the control unit, the processing unit receiving the input signal, the speech correlates and the control signal and processing the speech correlates according to the control signal for extracting speech from the input signal.

22. The noise reduction unit of claim 21, wherein the processing unit processes the input signal by selecting appropriate speech correlates based on the environmental correlates and tracking the appropriate speech correlates.

23. The noise reduction unit of claim 22, wherein the processing unit employs one of a Kalman filter and a particle filter for tracking the appropriate speech correlates.

24. A compensator for compensating for hearing loss in a hearing-aid, the compensator comprising:

a) a normal hearing model unit for receiving an input signal and generating a normal hearing signal;

b) a neuro-compensator unit for receiving the input signal and providing a pre-processed signal by applying a set of weights to the input signal;

c) a damaged hearing model unit connected to the neuro-compensator unit for receiving the pre-processed signal and providing an impaired hearing signal; and,

d) a comparison unit connected to the normal hearing model unit and the damaged hearing model unit for generating an error signal based on a comparison of the normal hearing signal and the impaired hearing signal; wherein, the error signal is provided to the neuro-compensator unit for adjusting the set of weights such that the normal hearing signal and the impaired hearing signal are substantially similar.

25. The compensator of claim 24, wherein the neuro-compensator is a neural network.

26. The compensator of claim 25, wherein the neuro-compensator applies a set of gain coefficients to the input signal, each gain coefficient being defined for a particular frequency band i according to $G_i = \dfrac{v_i f_i^2}{\sum_j w_j f_j^2 + \sigma}$ where f_i^2 is energy at frequency band i, w_j is a weight at frequency band j and σ is a constant related to the energy f_i^2.

27. The compensator of claim 25, wherein a weight W_j from the set of weights is defined for a particular time-slice at the i^{th} frequency according to

$W_i = \dfrac{v_i}{\left(\sum_{j=1}^{20} w_{ij} f_j\right)^{1/4} + \left[\sum_{k=0}^{4}\left(z_{ik} \sum_{j=1}^{20} f_j^{n-k}\right)^{1/4}\right] + \sigma}$ where f_j is the magnitude of the input signal in the j^{th} frequency band, v_i is optimized average gain, w_{ii} is optimized band to band inhibition, z_{ik} is optimized total power inhibition for past times and σ is a constant.

28. The compensator of claim 24, wherein the error signal is defined according to a Neural Articulation Index (NAI) of the form $NAI = \sum_{i=1}^{N} \alpha_i \cdot ND_i$ where N is a number of frequency bands, α_i is a weight for frequency band i, and ND (Neural Distortion) is defined by $ND = 1 - \dfrac{Test \cdot Control'}{Control \cdot Control'}$ where Test is a vector of instantaneous spiking rates provided by the damaged hearing model unit and Control is a vector of instantaneous spiking rates provided by the normal hearing model unit.

29. A method of processing an acoustic input signal and providing at least one output acoustic signal to a user of a hearing-aid system, the method comprising providing a first channel and a second channel, wherein one of channels includes an adaptive delay, and for the first channel, the method comprises:

a) providing directional processing to the acoustic input signal for generating a first directional signal;

b) processing the first directional signal for providing a first noise reduced signal by utilizing correlative measures for identifying a speech signal of interest in the first directional signal; and,

c) processing the first noise reduced signal for providing a first compensated signal for compensating for a hearing loss of the user.

30. The method of claim 29, wherein for the second channel the method includes:

d) providing directional processing to the acoustic input signal for generating a second directional signal;

e) processing the second directional signal for providing a second noise reduced signal by utilizing correlative measures for identifying a speech signal of interest in the second directional signal; and,

f) processing the second noise reduced signal for providing a second compensated signal for compensating for a hearing loss of the user.

31. The method of claim 30, wherein the method further comprises providing an appropriate delay to one of the first compensated signal and the second compensated signal for matching processing delay in the first and second channels.

32. The method of claim 29, wherein the method further comprises utilizing atomic decomposition phonemic processing for generating the correlative measures.

33. The method of claim 32, wherein the atomic decomposition phonemic processing comprises mapping a portion of the first directional signal into a five-dimensional space which comprises dimensions of: duration in time, duration in frequency, temporal centers of gravity, spectral centers of gravity, and change of spectral centers of gravity.

34. The method of claim 33, wherein the mapping is performed according to: $h_{T_c,F_c,\sigma_T,\sigma_F,\beta}(t,f)=\frac{1}{2\pi\sigma_T^2\sigma_F^2}e^{-\left[\frac{1}{2(1-\beta^2)}\left(\frac{(t-T_c)^2}{\sigma_T^2}-\frac{2\beta(t-T_c)(f-F_c)}{\sigma_T\sigma_F}+\frac{(f-F_c)^2}{\sigma_F^2}\right)\right]}$.

35. The method of claim 32, wherein the atomic decomposition phonemic processing comprises correlating an atom with a portion of the first directional signal according to: $\gamma_p=\arg\max_{\gamma}\left|\left\langle s_{p-1}(t), f\left(\sigma_T,\sigma_F\right)h_{\gamma}(t)\right\rangle\right|^2$.

36. The method of claim 29, wherein the method further comprises providing acoustic correlative tracking for generating the correlative measures, wherein the acoustic correlative tracking comprises:

d) receiving an input signal and generating a plurality of speech and environment correlates;

e) receiving the speech correlates and the environment correlates and generating a control signal; and,

f) processing the speech correlates according to the control signal for extracting speech from the input signal.

37. The method of claim 36, wherein processing the speech correlates includes selecting appropriate speech correlates based on the environmental correlates and tracking the appropriate speech correlates.

38. The method of claim 29, wherein step (c) comprises:

d) receiving an input signal and generating a normal hearing signal based on a normal hearing model;

e) receiving the input signal and providing a pre-processed signal by applying a set of weights to the input signal;

f) receiving the pre-processed signal and providing an impaired hearing signal based on an impaired hearing model; and,

g) generating an error signal based on a comparison of the normal hearing signal and the impaired hearing signal;

wherein, the error signal is used to adjust the set of weights such that the normal hearing signal and the impaired hearing signal are substantially similar.

39. The method of claim 38, wherein applying the set of weights results in applying a set of gain coefficients to the input signal, each gain coefficient being defined for a particular frequency band i according to $G_i = \frac{v_i f_i^2}{\sum_j w_j f_j^2 + \sigma}$

where f_i^2 is energy at frequency band i, w_j is a weight at frequency band j and σ is a constant related to the energy f_i^2.

40. The method of claim 38, wherein a weight W_j from the set of weights is defined for a particular time-slice at the i^{th} frequency band according to $W_i = \dfrac{v_i}{\left(\sum_{j=1}^{20} w_{ij} f_j\right)^{1/4} + \left[\sum_{k=0}^{4}\left(z_{ik}\sum_{j=1}^{20} f_j^{n-k}\right)^{1/4}\right] + \sigma}$ where f_j is the magnitude of the input signal in the j^{th} frequency band, v_i is optimized average gain, w_{ii} is optimized band to band inhibition, z_{ik} is optimized total power inhibition for past times and σ is a constant.

41. The method of claim 38, wherein the error signal is defined according to a Neural Articulation Index (NAI) of the form $NAI = \sum_{i=1}^{N} \alpha_i \cdot ND_i$ where N is a number of frequency bands., α_i is a weight for frequency band i, and ND (Neural Distortion) is defined by $ND = 1 - \frac{Test \cdot Control'}{Control \cdot Control'}$ where Test is a vector of instantaneous spiking rates generated by the damaged hearing model and Control is a vector of instantaneous spiking rates provided by the normal hearing model.

42. A method of reducing noise in an input signal and generating a noise reduced signal for a hearing aid, the method comprising:

a) generating correlative measures for identifying a speech signal of interest in the input signal; and,

b) tracking the speech signal of interest to produce the noise reduced signal.

43. The method of claim 42, wherein the method employs atomic decomposition phonemic processing to provide the correlative measures.

44. The method of claim 43, wherein the atomic decomposition phonemic processing comprises mapping a portion of the first directional signal into a

five-dimensional space which comprises dimensions of: duration in time, duration in frequency, temporal centers of gravity, spectral centers of gravity, and change of spectral centers of gravity.

45. The method of claim 44, wherein the mapping is performed according to: $h_{T_c,F_c,\sigma_T,\sigma_F,\beta}(t,f)=\frac{1}{2\pi\sigma_T^2\sigma_F^2}e^{-\left[\frac{1}{2(1-\beta^2)}\left(\frac{(t-T_c)^2}{\sigma_T^2}-\frac{2\beta(t-T_c)(f-F_c)}{\sigma_T\sigma_F}+\frac{(f-F_c)^2}{\sigma_F^2}\right)\right]}$.

46. The method of claim 43, wherein the atomic decomposition phonemic processing comprises correlating an atom with a portion of the input signal according to: $\gamma_p = \arg\max_{\gamma}\left|\left\langle s_{p-1}(t), f\left(\sigma_T,\sigma_F\right)h_\gamma(t)\right\rangle\right|^2$.

47. The method of claim 42, wherein providing the correlative measures includes:

c) generating a plurality of speech and environment correlates;

d) generating a control signal based on the speech correlates and the environment correlates; and,

e) processing the speech correlates according to the control signal for extracting speech from the input signal.

48. The method of claim 47, wherein the processing of step (e) includes selecting appropriate speech correlates based on the environmental correlates.

49. A method of compensating for hearing loss in a hearing-aid, the method comprising:

a) receiving an input signal and generating a normal hearing signal based on a normal hearing model;

b) receiving the input signal and providing a pre-processed signal by applying a set of weights to the input signal;

c) receiving the pre-processed signal and providing an impaired hearing signal based on an impaired hearing model; and,

d) generating an error signal based on a comparison of the normal hearing signal and the impaired hearing signal;

wherein, the error signal is used to adjust the set of weights such that the normal hearing signal and the impaired hearing signal are substantially similar.

50. The method of claim 49, wherein applying the set of weights results in applying a set of gain coefficients to the input signal, each gain coefficient being defined for a particular frequency band i according to $G_i = \frac{v_i f_i^2}{\sum_j w_j f_j^2 + \sigma}$ where f_i^2 is energy at frequency band i, w_j is a weight at frequency band j and σ is a constant related to the energy f_i^2.

51. The method of claim 49, wherein a weight W_j from the set of weights is defined for a particular time-slice at the i^{th} frequency band according to $W_i = \frac{v_i}{\left(\sum_{j=1}^{20} w_{ij} f_j\right)^{1/4} + \left[\sum_{k=0}^{4}\left(z_{ik} \sum_{j=1}^{20} f_j^{n-k}\right)^{1/4}\right] + \sigma}$ where f_j is the magnitude of the input signal in the j^{th} frequency band, v_i is optimized average gain, w_{ii} is optimized band to band inhibition, z_{ik} is optimized total power inhibition for past times and σ is a constant.

52. The method of claim 49, wherein the error signal is defined according to a Neural Articulation Index (NAI) of the form $NAI = \sum_{i=1}^{N} \alpha_i \cdot ND_i$ where N is a number of frequency bands, α_i is a weight for frequency band i, and ND (Neural Distortion) is defined by $ND = 1 - \frac{Test \cdot Control'}{Control \cdot Control'}$ where Test is a vector of instantaneous spiking rates provided by the damaged hearing model and Control is a vector of instantaneous spiking rates provided by the normal hearing model.